

UNITED STATES PATENT APPLICATION  
FOR  
SCALE FACTOR BASED BIT SHIFTING IN FINE GRANULARITY SCALABILITY  
AUDIO CODING  
BY  
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## **DESCRIPTION OF THE INVENTION**

### **Field of the Invention**

[001] The present invention generally relates to audio coding and, more particularly, to scale factor based bit shifting (SFBBS) in fine granularity scalability (FGS) audio coding.

### **Background of the Invention**

[002] Fine granularity scalability (FGS) includes a multitude of audio coding applications such as real-time multimedia streaming and dynamic multimedia storage. In particular, FGS has been adopted by the Motion Picture Experts Group (MPEG) and incorporated into the MPEG 4 international standard, including AAC.

[003] In conventional coding such as AAC in MPEG-4, first codes of the information are used in left and right channels at a place of the header in processing audio signals. The left-channel data are coded and the right-channel data are then coded. That is, coding is processed in the order of the header, left and right channels. When information for the left and right channels are arranged and transmitted irrespective of significance after the header is processed in such a manner, signals for the right channel positioned backwards will disappear first if the bit rate is lowered. The transmission performance will seriously degrade as a result.

[004] In FGS audio coding, a base layer and an enhancement layer are transmitted. The single enhancement layer, after quantization of the data therein, is transmitted with varied bit rates. Truncation of the quantized data also takes place as layer size limits are applied in the enhancement layer. Noise shaping is

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implemented to minimize quantization noise, under a masking level so it will be imperceptible to the human ear. For noise shaping, psychoacoustics are applied to control errors in the quantization process with scale factors being associated with a plurality of sub-bands. The most important characteristics of human acoustics in coding a digital audio signal include a masking effect (as an audio signal is inaudible due to another signal) and a critical band feature (as noises having the same amplitude are differently perceived when the noise signal is within or without a critical band). These characteristics are utilized so the range of noise allocated within a critical band is calculated in generating quantization noise corresponding to the calculated range to minimize data loss due to the coding. However, errors introduced by the disposal of the truncated data are not governed by the psychoacoustic model.

[005] There is thus a general need in the art for a method and system of audio coding to overcome at least the aforementioned shortcomings in the art. A particular need exists in the art for an optimal method and system in audio coding overcoming performance degradation issues when information in channels are arranged and transmitted irrespective of significance as the bit rate is lowered. A further need exists in the art for an optimal FGS method and system in audio coding overcoming the limitations of the psychoacoustic model in controlling errors in truncation of quantized data.

### **SUMMARY OF THE INVENTION**

[006] Accordingly, one embodiment of the present invention is directed to a scale factor based bit shifting (SFBBS) method and system in FGS audio coding that

obviate one or more of the problems due to limitations and disadvantages of the related art.

[007] To achieve these and other advantages, as the audio signals are quantized in an order of most significant bits (MSBs) to least significant bits (LSBs), the significance of the MSBs is increased with respect to the LSBs. In the plurality of sub-bands in which the audio signals are quantized, the MSBs are shifted upwards in terms of significance by the respective scale factors assigned thereto by the psychoacoustic model. Scale factors correspond to the noise tolerance in each of the sub-bands. The sub-bands with less error tolerance are generally associated with larger scale factors. Small error tolerance means that the human ear will be more sensitive to the frequency range defined by the sub-band corresponding to that small error tolerance. That is, if the error tolerance is small in a sub-band, the quantized data in that sub-band are more significant as they must be more sensitive to the human ear. If the scale factor in a particular sub-band exceeds a threshold value, the quantized data in that sub-band are shifted by the respective scale factor, i.e., the bits in that sub-band are shifted upwards by the same number of significance levels as the value of the sub-band's scale factor.

[008] In accordance with the purpose of the invention as generally embodied and broadly described, there is provided a scale factor based bit shifting (SFBBS) processor processing audio signals in an order of most significant bits to least significant bits that includes a psychoacoustic model determining a plurality of scale factors corresponding to a plurality of spectral sub-bands according to respective noise tolerance of each of the sub-bands, a bit shifter shifting the processed audio

signals in the spectral sub-bands by the respective scale factors if they exceed a threshold value, and a bit slicer coding and truncating the processed audio signals.

[009] In another aspect, the SFBBS processor according to the invention further comprises a quantizer quantizing the processed audio signals. Such SFBBS processor can be implemented in MPEG AAC.

[010] In yet another aspect, the SFBBS processor according to the invention further comprises a quantizer and de-quantizer respectively quantizing and de-quantizing the processed audio signals, and a subtractor taking a difference between the and the de-quantized audio signals. Such SFBBS processor can be implemented in MPEG-4 bit slice arithmetic coding (BSAC).

[011] There is also provided a method for processing audio signals comprising the steps of quantizing the audio signals in spectral lines into quantized data in a plurality of sub-bands in an order of most significant bits to least significant bits, determining a plurality of scale factors corresponding to each of the sub-bands according to respective noise tolerance of each of the sub-bands, bit shifting the quantized data by the respective scale factors if they exceed a threshold value, coding the quantized data, truncating the quantized data, de-shifting the coded data with the respective scale factors, de-quantizing the coded data, and decoding the coded data.

[012] According to a further embodiment according to the present invention, there is provided a method for coding audio signals in a base layer and an enhancement layer comprising the steps of quantizing the audio signals in spectral lines into quantized data in a plurality of sub-bands in an order of most significant

bits to least significant bits, determining a plurality of scale factors corresponding to each of the sub-bands according to respective noise tolerance of each of the sub-bands, bit shifting the quantized data by the respective scale factors if they exceed a threshold value, coding the quantized data in the base layer, coding the quantized data in the enhancement layer, truncating the quantized data in the enhancement layer up to respective layer size limits, de-shifting the coded data with the respective scale factors, de-quantizing the coded data, and decoding the coded data.

[013] In one aspect, the method according to the invention is implemented in MPEG additive arithmetic coding (AAC) or MPEG-4 bit slice arithmetic coding (BSAC).

[014] In another aspect, the method according to the invention utilizes Huffman coding, run length (RL) coding or arithmetic coding (AC), e.g., in an MPEG 4 AAC system having an AAC encoder and AAC decoder.

[015] In an additional aspect, the method according to the invention further comprises the steps of amplifying the coded data with the respective scale factors, and de-amplifying the decoded data with the respective scale factors.

[016] Further in accordance with another embodiment, there is provided an SFBBS structure having an encoder and decoder for coding and transmitting a base layer and an enhancement layer according to the present invention. Since most of the errors are generated during quantization, a de-quantizer is advantageously provided in the encoder and the difference of the data being coded is taken before and after quantization. As the SFBBS are performed, the single enhancement layer is accordingly constructed.

[017] An exemplary encoder in an SFBBS structure according to one embodiment of the present invention primarily comprises a psychoacoustic model, filter, quantizer, noiseless coder, subtractor, de-quantizer, shifter and bit slicer. A decoder of an additive SFBBS structure according to the present invention primarily comprises a scale factor decoder, spectrum decoder, de-quantizer, adder, filter, de-shifter and bitmap decoder.

[018] In one aspect, the SFBBS structure according to the invention is implemented in MPEG AAC or MPEG-4 BSAC.

[019] A scale factor based bit shifting (SFBBS) system in an additive fine granularity scalability (FGS) structure according to the present invention comprises an encoder including a quantizer quantizing the audio signals in spectral lines into quantized data and errors in a plurality of sub-bands in an order of most significant bits to least significant bits, a psychoacoustic model determining a plurality of scale factors corresponding to each of the sub-bands according to respective noise tolerance of each of the sub-bands, a coder coding the quantized data in the base layer, a de-quantizer de-quantizing the quantized data, a subtractor taking a difference of the quantized data and the de-quantized data, a bit shifter shifting the difference between the quantized and de-quantized data in the sub-bands by the respective scale factors if they exceed a threshold value, and a bit slicer coding the and truncating the difference between the quantized and de-quantized data. The system according to this particular embodiment of the present invention further comprises a decoder having a scale factor decoder decoding the scale factors, a spectrum decoder decoding the quantized data, a de-quantizer de-quantizing the

quantized data, a de-shifter de-shifting the coded data, and a decoder decoding the coded data.

[020] In a further aspect, an SFBBS system is further provided for implementation with bit slice arithmetic coding (BSAC) in MPEG-4.

[021] A particular advantage of the present invention is that no further information will need to be sent in the enhancement layer, advantageously avoiding bandwidth issues and additional overhead as the audio signal quality is optimized by as much as 3 decibels. As the scale factors are utilized in SFBBS, the present invention is wholly scalable and compatible with FGS audio systems.

[022] Additional objects and advantages of the invention will be set forth in part in the description which follows, and in part will be obvious from the description, or may be learned by practice of the invention. The objects and advantages of the invention will be realized and attained by means of the elements and combinations particularly pointed out in the appended claims.

[023] It is to be understood that both the foregoing general description and the following detailed description are exemplary and explanatory only and are not restrictive of the invention as claimed.

[024] The accompanying drawings, which are incorporated in and constitute a part of this specification, illustrate several embodiments of the invention and together with the description, serve to explain the principles of the invention.

### **BRIEF DESCRIPTION OF THE DRAWINGS**

[025] Fig. 1 is a flow diagram exemplarily illustrating a communications method according to an embodiment of the present invention;



[026] Fig. 2 is a spectral diagram exemplarily illustrating the scale factor based bit shifting (SFBBS) according to the present invention;

[027] Figs. 3 and 4 are diagrams illustrating an encoder and decoder of an additive SFBBS structure in accordance with the present invention; and

[028] Figs. 5 and 6 are block diagrams respectively illustrating an exemplary BSAC encoder and decoder with scale factor based bit shifting (SFBBS) according to yet another embodiment of the present invention.

### **DESCRIPTION OF THE EMBODIMENTS**

[029] Reference will now be made in detail to the present embodiment of the invention, an example of which is illustrated in the accompanying drawings.

Wherever possible, the same reference numbers will be used throughout the drawings to refer to the same or like parts.

[030] Fig. 1 is a flow diagram of a communications method according to one embodiment of the present invention. Referring to Fig. 1, there is provided a method for coding audio signals in a base layer and an enhancement layer comprising the steps of quantizing the audio signals in spectral lines into quantized data in a plurality of sub-bands in an order of most significant bits to least significant bits (step 101), determining a plurality of scale factors corresponding to each of the sub-bands according to respective noise tolerance of each of the sub-bands (step 102), bit shifting the quantized data by the respective scale factors if they exceed a threshold value (step 103), coding the quantized data in the base layer (step 104) and the enhancement layer (step 105), truncating the quantized data in the enhancement layer up to respective layer size limits (step 106), de-shifting the coded data with the

respective scale factors (step 107), de-quantizing the coded data (step 108), and decoding the coded data (step 109). In one aspect, the method according to this particular embodiment is advantageously implemented in MPEG-4 BSAC.

[031] In another aspect, the method according to the present invention utilizes Huffman coding, run length (RL) coding or arithmetic coding (AC).

[032] In yet another aspect, the method according to the present invention further comprises the steps of converting the audio signals from a time domain to a frequency domain, e.g., through modified discrete cosine transform (MDCT), and converting the decoded data from the frequency domain to the time domain by IMDCT.

[033] In an additional aspect, the method according to the invention further comprises the steps of amplifying the coded data with the respective scale factors, and de-amplifying the decoded data with the respective scale factors.

[034] As the audio signals are quantized in an order of most significant bits (MSBs) to least significant bits (LSBs), a particular advantage of the present invention is that the significance of the MSBs is increased with respect to the LSBs. In the plurality of sub-bands in which the audio signals are quantized, the MSBs are shifted upwards in terms of significance by the respective scale factors assigned thereto by the psychoacoustic model.

[035] The method according to a further embodiment of the invention advantageously comprises quantizing the audio signals in spectral lines into quantized data in a plurality of sub-bands in an order of most significant bits to least significant bits, determining a plurality of scale factors corresponding to each of the

sub-bands according to respective noise tolerance of each of the sub-bands, de-quantizing the quantized data, bit shifting the difference between the quantized and de-quantized data in the sub-bands by the respective scale factors if they exceed a threshold value, coding and truncating the quantized difference. In one aspect, the method according to this particular embodiment is implemented in MPEG AAC.

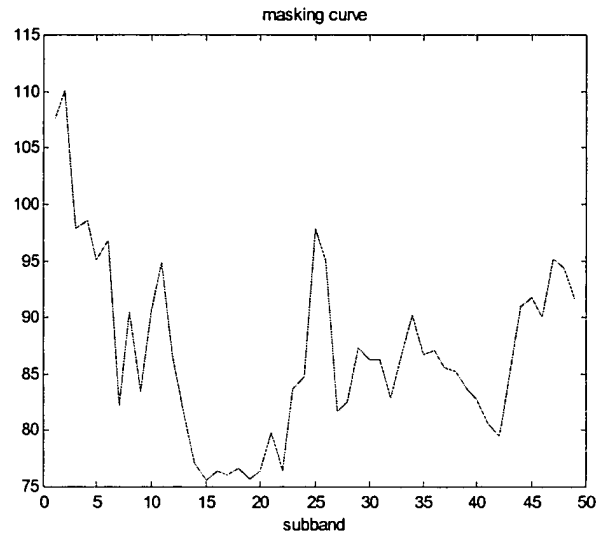
[036] Fig. 2 is a spectral diagram exemplarily illustrating the scale factor based bit shifting (SFBBS) according to the present invention. Scale factors correspond to the noise tolerance in each of the sub-bands  $i, i+1, i+2, \dots$  in their respective spectral energy. The sub-bands with less error tolerance are generally associated with larger scale factors. Small error tolerance means that the human ear will be more sensitive to the frequency range defined by the sub-band corresponding to that small error tolerance. That is, if the error tolerance is small in a sub-band, the quantized data in that sub-band are more significant as they must be more sensitive to the human ear. If the scale factor in a particular sub-band exceeds a threshold value, the quantized data in that sub-band are shifted by the respective scale factor, i.e., the bits in that sub-band are shifted upwards by the same number of significance levels as the value of the sub-band's scale

factor.

band	scalefactor	band	scalefactor
1	0	25	0
2	0	26	0
3	0	27	1
4	0	28	1
5	0	29	0
6	1	30	0
7	2	31	0
8	0	32	1
9	0	33	0
10	0	34	0
11	1	35	0
12	0	36	0
13	0	37	0
14	3	38	0
15	1	39	0
16	3	40	X
17	4	41	X
18	3	42	1
19	2	43	X
20	3	44	X
21	1	45	X
22	3	46	0
23	0	47	0
24	1	48	X
		49	X

[037] **Table A**

**Table B**



[038] The above Tables A and B exemplarily illustrate the relationship between a plurality of scale factors and the masking curve of a single MPEG-4 AAC coded frame in tabular and graphical forms, respectively. At the sub-bands where the masking level is smaller, the values of their respective scale factors are higher. The present invention advantageously exploits this relationship in scale factor-based bit shifting (SFBBS) in optimizing the decoded audio signal quality at low bit rates.

[039] Accordingly, the invention generally provides a scale factor based bit shifting (SFBBS) processor processing audio signals in an order of most significant bits to least significant bits that includes a psychoacoustic model determining a plurality of scale factors corresponding to a plurality of spectral sub-bands according

to respective noise tolerance of each of the sub-bands, a bit shifter shifting the processed audio signals in the spectral sub-bands by the respective scale factors if they exceed a threshold value, and a bit slicer coding and truncating the processed audio signals.

[040] In another aspect, the SFBBS processor according to the invention further comprises a quantizer quantizing the processed audio signals. Such SFBBS processor can be implemented in MPEG AAC.

[041] In yet another aspect, the SFBBS processor according to the invention further comprises a quantizer and de-quantizer respectively quantizing and de-quantizing the processed audio signals, and a subtractor taking a difference between the and the de-quantized audio signals. Such SFBBS processor can be implemented in MPEG-4 bit slice arithmetic coding (BSAC).

[042] Referring again to Fig. 2, for example, sub-band (i+2) is a sub-band with low noise tolerance with a corresponding high scale factor. If the scale factor of the sub-band is 4, all bit values in the spectral lines in the sub-band are shifted upwards by 4 energy levels (as exemplarily shown in Fig. 2). Once these more significant bits are shifted, they are accordingly placed in the more important sub-bands (i.e., those with less error tolerance) closer to the beginning of the enhancement layer. After the bit shifting, some or all of the least significant bit values in the spectral lines are not coded or discarded, advantageously saving valuable bandwidth.

[043] For high bit rate audio coding, the coding errors are kept under a masking level so they are imperceptible to the human ear. However, for low bit rate

coding, the errors are still perceptible. Psychoacoustics are used in the encoder to minimize the perceptible errors. For a given bit rate, a psychoacoustic model is used in the encoder to best shape the noise level. The same noise shaping issue is encountered when an enhancement layer or parts thereof are added or improved, which is akin to changing the bit rate in the bit stream. It will be impractical if the bit rate allocation algorithm is recursively applied, since the actual bit rate for the received data in an enhancement layer cannot be foreseen by the encoder. The present invention advantageously utilizes psychoacoustics in noise shaping the coded data while optimizing the performance of the FGS enhancement layer. Even though the actual bit rate as seen by the decoder is not known to the encoder, the encoder can still perform noise shaping psychoacoustically, using scale factor-based bit shifting or SFBBS.

[044] The methodology according to the invention can be described and iteratively expressed in an inner loop and an outer loop. An exemplary pseudo code expression for the inner loop is shown in Table C as follows:

```
if (counted_bits > available_bits) then
    common_scalefac = common_scalefac + quantizer_change
else
    common_scalefac = common_scalefac - quantizer_change
end if
```

[045] According to Table C, a common scale factor is determined by comparing the number of counted bits and available bits. If the number of counted bits is greater than the available bits, the common scale factor is increased by a positive quantization change. Conversely, if the number of counted bits is not greater than the available bits, the common scale factor is decreased by the quantization change.

[046] An outer loop is used to determine the respective scale factor for each of the sub-bands. An exemplary pseudo code expression for the outer loop is shown in Table D as follows:

```

do for each scalefactor band sb:
    error_energy(sb)=0
    do from lower index to upper index i of scalefactor band
        error_energy(sb) = error_energy(sb) + (abs( mdct_line(i)
            - (x_quant(i) 4/3 * 2(-1/4 * (scalefactor(sb) - common_scalefac )))))2
    end do
end do

do for each scalefactor band sb
    if ( error_energy(sb) > xmin(sb) ) then
        scalefactor(sb) = scalefactor(sb) + 1
    end if
end do

```

[047] According to Table D, the error energy for each of the sub-bands is determined by taking the value of the original spectral energy level, e.g., through modified discrete cosine transform or MDCT, and adjusting it with de-quantization of the difference of the common scale factor and band scale factor values. Adjustment is made to the respective scale factor (i.e., incrementally by one) for each of the sub-bands if the error energy for the sub-band is greater than a threshold value.

[048] Figs. 3 and 4 are diagrams illustrating an encoder and decoder of an additive SFBBS structure in accordance with the present invention. Since most of the errors are generated during quantization, a de-quantizer is advantageously provided in the encoder and the difference of the data being coded is taken before and after quantization. As the SFBBS and bit slice are performed, the single enhancement layer is accordingly constructed. In one aspect, this additive SFBBS structure is advantageously implemented in MPEG AAC.

[049] For an additive fine granularity scalability (FGS) coding structure, there is provided a method comprising the steps of quantizing the audio signals in spectral lines into quantized data and errors in a plurality of sub-bands in an order of most significant bits to least significant bits, determining a plurality of scale factors corresponding to each of the sub-bands according to respective noise tolerance of each of the sub-bands, bit shifting the quantized errors by the respective scale factors if they exceed a threshold value, coding the quantized data in the base layer, coding the quantized data in the enhancement layer, truncating the quantized data in the enhancement layer up to respective layer size limits, de-shifting the coded data with the respective scale factors, de-quantizing the coded data, and decoding the coded data.

[050] Referring to Fig. 3, an encoder of an additive SFBBS structure for coding and transmitting a base layer and an enhancement layer according to the present invention comprises a psychoacoustic model 301, filter 302, quantizer 303, noiseless coder 304, subtractor 305, de-quantizer 306, shifter 307 and bit slicer 308. The original audio signals are input into the encoder at psychoacoustic model 301 and filter 302. Filter 302 converts the input audio signals from the time domain to signals in the frequency domain for further processing. Psychoacoustic model 301 couples the frequency-domain signals converted by filter 302 by signals of sub-bands corresponding to scale factors. A masking threshold at each sub-band is calculated using a masking phenomenon generated by interaction with the respective signals. Quantizer 303 quantizes the frequency-domain signals with respect to their spectral energy and their respective noise tolerance in a plurality of



sub-bands. De-quantizer 306 is provided in the encoder and the difference of the data being coded is taken at subtractor 305 before and after quantization at quantizer 303. At the shifter 307, the quantized errors for the plurality of sub-bands are bit shifted by the respective scale factors if they exceed a threshold value. After bit slicing at the slicer 308, the single enhancement layer is coded and accordingly constructed. For bit slicing, instead of vertically sending the bits in the order of each word, the bits are horizontally sent in the order of each bit slice according to its significance in the respective bit array. After coding the enhancement layer, the bits with greater significance will be placed closer to the beginning of the enhancement layer. After noiseless coding in the coder 304, the base layer is coded and accordingly constructed.

[051] A particular advantage is when only a part of the enhancement layer is received, the decoder of an additive SFBBS structure according to the present invention still will have the general shape of the entire spectrum, even though some of the details may have been lost. Advantageously according to the present invention, it will not matter at which point the enhancement layer is truncated, the received data will be decodable as long as they are received generally without error. The longer the enhancement layer is received at the decoder, the more detail can be decoded by the decoder, which in turn leads to superior audio signal quality.

[052] After the quantization error is received, bit slicing is performed in bit slicer 308, after at least some of the bits have been shifted at shifter 307. The significance of bits that are originally less significant is increased as their respective position is moved toward the beginning of the enhancement layer and have them

sent earlier. For shifting for the best performance, scale factors are utilized as the noise level is accordingly reshaped for each extra bit received from the enhancement layer. As the scale factors are received in the decoder, there is advantageously no need to send any extra information in the enhancement layer.

[053] Referring to Fig. 4, a decoder of an additive SFBBS structure according to the present invention comprises a scale factor decoder 401, spectrum decoder 402, de-quantizer 403, adder 404, filter 405, de-shifter 406 and bitmap decoder 407. At the decoder 401, the coded data in the base layer and the corresponding scale factors are decoded. The coded data and their respective spectral lines are decoded at the spectrum decoder 402 and their respective spectral energy de-quantized at the de-quantizer 403. The coded data in the enhancement layer are de-shifted by the respective scale factors in the sub-bands at de-shifter 406. After decoding at bitmap decoder 407, the decoded data are forwarded to adder 404 to accordingly construct the audio signals. The decoded audio signals are then converted from the frequency domain to the time domain at filter 405.

[054] In one aspect, the present invention utilizes Huffman coding, run length (RL) coding or arithmetic coding (AC), e.g., in an MPEG-4 system with a bit slice arithmetic coder (BSAC). Figs. 5 and 6 are block diagrams respectively illustrating an exemplary BSAC encoder and decoder in a structure embedded with scale factor based bit shifting (SFBBS) according to yet another embodiment of the present invention. In one aspect, this embedded structure is advantageously implemented in MPEG-4 BSAC.

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[055] Accordingly, the encoder comprises a filter 502, psychoacoustic model 501, temporal noise shaper or TNS 503, prediction modules 504, 506 and 507, intensity processor 505, M/S processor 508, quantizer 509, SFBBS shifter 510 and bit slice arithmetic coder 511. Filter 502 converts input audio signals from a time domain to a frequency domain. Psychoacoustic model 501 couples the frequency-domain signals converted by filter 502 by signals of sub-bands corresponding to scale factors. A masking threshold at each sub-band is calculated using a masking phenomenon generated by interaction with the respective signals. TNS 503, optionally used in the encoder, controls the temporal noise shape of the quantization noise within each window for signal conversion, which can be temporally shaped by filtering frequency data. Intensity processor 505, also optionally used in the encoder, encodes only the quantized information for the sub-band of one of two channels with the sub-band of the other channel being transmitted. Prediction modules 504, 506 and 507, optionally used in the encoder, estimate frequency coefficients of the current frames. The difference of the predicted values and the actual frequency components is quantized and coded in effectively reducing the quantity of generated usable bits. M/S processor 508, optionally used in the encoder, respectively converts a left-channel signal and a right-channel signal into additive and subtractive signals of two signals, to then process the same. Quantizer 509 scalar-quantizes the frequency signals of each of the sub-bands so the magnitude of the quantization noise of each sub-band is smaller than the masking threshold in ensuring imperceptibility to the human ear. At SFBBS shifter 510, the quantized data for the plurality of sub-bands are bit shifted by the respective scale factors if they exceed a

threshold value, as set forth herein according to the principles of the present invention. At bit slice arithmetic coder 511, the quantized frequency data are coded by combining the side information (including scale factors) of the corresponding sub-band and the quantization information of audio data. Quantized data are sequentially coded in the order ranging from the most significant bit (MSB) sequences to the least significant bit (LSB) sequences, and from the lower frequency components to the higher frequency components. Left and right channels are alternately coded in vectors to perform coding of a base layer. After the base layer is coded, the side information (including scale factors) for the next enhancement layer and quantized data are coded so the thus-formed bit streams have a layered structure. Bit streams are then generated and multiplexed for transmission to the decoder.

[056] Referring to Fig. 6, the decoder in the embedded structure embodiment according to the present invention comprises a bit slice arithmetic decoder 601, SFBBS de-shifter 602, de-quantizer 603, M/S processor 604, prediction modules 605, 606 and 608, intensity processor 607, TNS 609 and filter 610. As the bit streams for the coded data are received and de-multiplexed, the header information and coded data are separated in the order of generation of the bit streams. Bit slice arithmetic decoder 601 decodes the side information (including scale factors) and bit sliced quantized data in the order of generation of the input bit streams. At SFBBS de-shifter 602, the coded data are de-shifted by the respective scale factors in the sub-bands in accordance with the principles of the present invention as set forth herein. At de-quantizer 603, the decoded data are de-

quantized. M/S processor 604 processes the sub-band corresponding to the M/S processing performed in the encoder. If estimation is performed in the encoder, prediction modules 605, 606 and 608 search the same values as the decoded data in the previous frame through estimation in the same manner as in the encoder. The predicted signal is added with a decoded and de-multiplexed difference signal in restoring the original frequency components. TNS 609 controls the temporal shape of quantization noise with each window for conversion from the frequency domain to the time domain. The decoded data are restored as temporal signals using a conventional audio algorithm such as AAC in MPEG-4. De-quantizer 603 restores the decoded scale factor and quantized data into signals having the original magnitudes. Filter 610 then converts the de-quantized signals into signals of a temporal domain.

[057] Other embodiments of the invention will be apparent to those skilled in the art from consideration of the specification and practice of the invention disclosed herein. It is intended that the specification and examples be considered as exemplary only, with a true scope and spirit of the invention being indicated by the following claims.